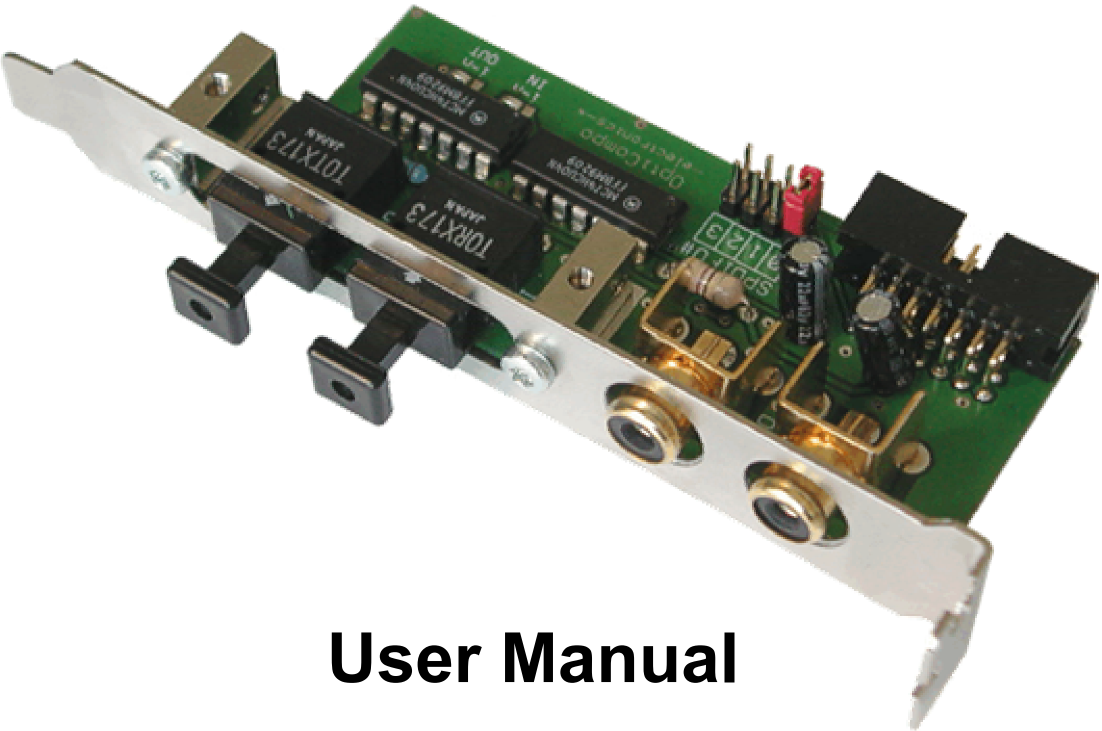
POCAB

Optical & Coaxial S/PDIF Adapter for SB Audigy, Live! (1024, Player, Value)



User Manual

http://www.opticompo.com

info@opticompo.com

optiCompo ELECTRONICS

2/15/1999 Ver. 1.0, 3/3/2000 Ver. 1.1, 08/20/2003 Ver 2.0

Thank you for purchasing optiCompo's *POCAB* (nice-*P*riced *O*ptical and *C*oaxial *A*dapter*B*oard).

Please read the following installation procedure carefully and you will enjoy a wide variety of SPDIF recording options with your sound card.

Some features your *POCAB* has to offer:

- **Interconnectivity** - Using the *POCAB* you are finally able to connect your S/PDIF audio devices to your SoundBlaster Audigy or Live!.
- **Independence** - You can interconnect S/PDIF audio devices independent of their output type - coaxial or optical jack.
- **Service** - If you have problems with your *POCAB*, please check our homepage for usefull hints and solutions:

  http://www.opticompo.com/

  Before contacting optiCompo, try to solve the problem by reading this User Manual or the manuals and help files for the other used devices and software. If these manuals leave any questions, we provide e-mail support at

  audiosupport@opticompo.com

So than: Just enjoy the digital world of music.

Your optiCompo team

Warning

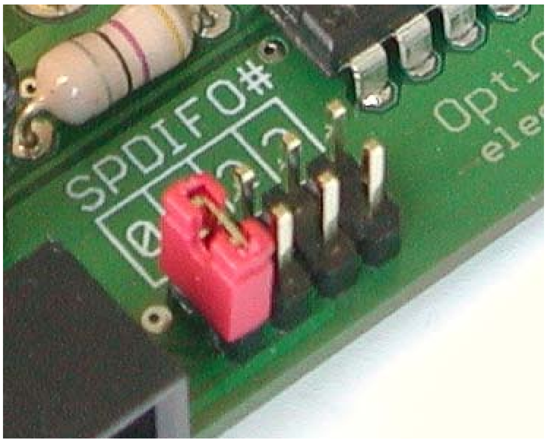
- Do not eat or pulverize any of the components of the *POCAB*, because some components contain toxic material!
- We are not liable for any damages caused by improper use of the *POCAB* or it's components!

****

Information in this manual is subject to change without notice.

Selecting the output signal source

The *POCAB* works basing on the digital signals interfaced by the SB Audigy/Live!
Please choose the SPDIF channel to be passed to the output by setting the jumper.



SPDIFO#0	front stereo channel, LF+RF (signal for 2 and 4 speaker mode)
SPDIFO#1	center and subwoofer channel
SPDIFO#2	mix of front and rear stereo channel
SPDIFO#3	stereo channel rear, LR+RR (only active in 4 speaker mode)

Default is SPDIFO#0.

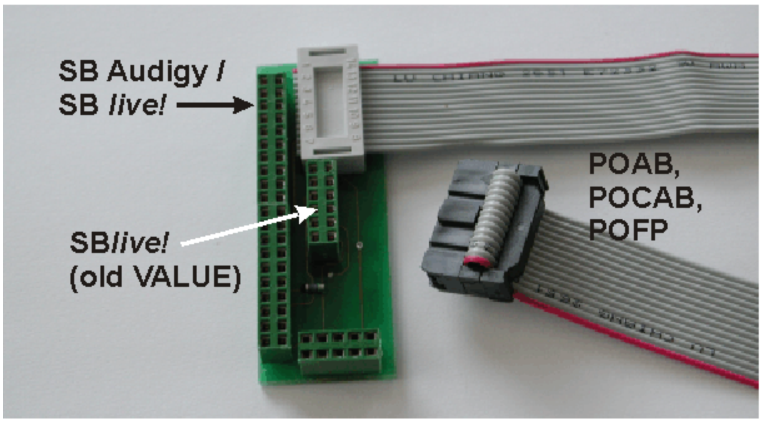
Installing the *POCAB*

1. Switch off your computer, remove the power supply cable and unscrew the computer casing.

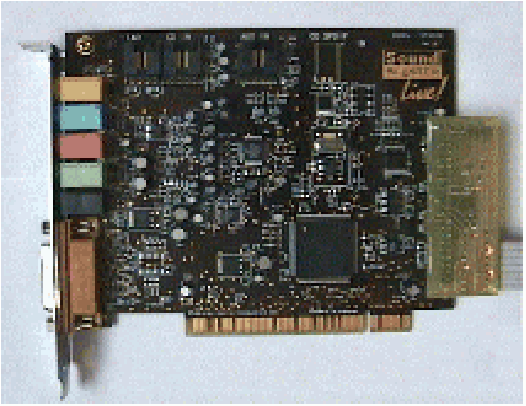
   ! Make sure to avoid electrostatical chargings (caused by clothes, carpets etc.); they can destroy electronic components.

2. Locate the sound card. The SBAudigy/Live! is a PCI card. In tower casings this usually means that it is installed upside down.
   In most cases you will have to remove the card to reach the used connectors.

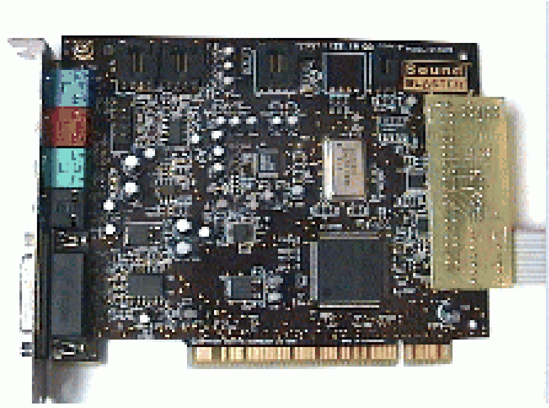
3. Up to now there are two supported SBAudigy/Live! extension pin-outs. The *POCAB* 's module cable has got a multi-adapter for the different hardware releases.



   The following pictures show how the small adapter is plugged to the SB Audigy/Live! card. In most cases you have to remove the card to reach the used connectors.



SB Audigy / Live!, Player, 1024 (Value) - 40 pin

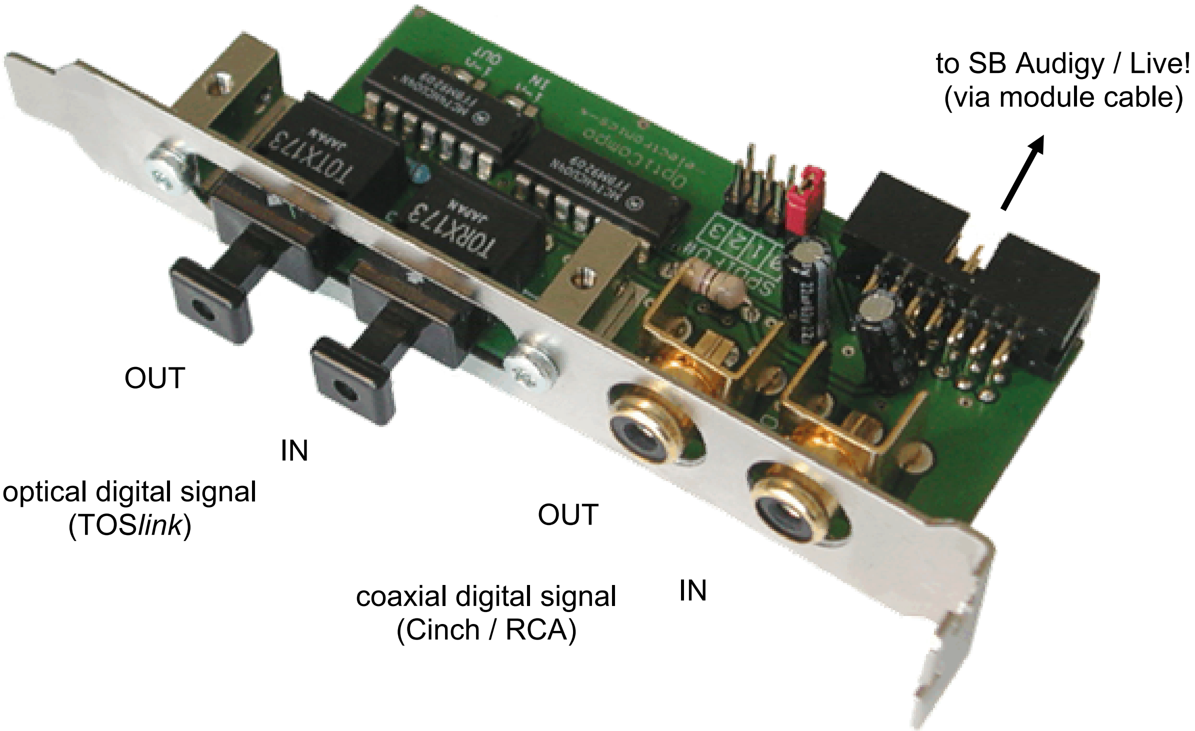


old "SB Live! Value" with 12pin connector

! During the (re)installation of the sound card you should take care that there is some space left between the multi-adapter and the next PCI / AGP card.

Although the adapter's back is coated with insulating material you should not risk a short circuit.

4. Plug the 14-pin connector of the module cable to the port of your *POCAB*.



5. Choose a free slot to install the *POCAB* module. Remove the slot cover and screw the *POCAB* tight in it's place.

! If possible the *POCAB* slot should be installed with a little distance to high-speed AGP or PCI cards, which may cause interferences.

6. Check for tight screws and a tight fit of the SB Audigy/Live!, close the casing and switch on the computer.

After switching on the computer there will be an immediate signal on SPDIF-OUT.

The easiest way to check this is to controll whether the optical output emitts red light (remove the protective cap).
This means the hardware is installed correctly, bravo!

! If the optical output does not emit light after power-on, switch off the computer immediately and check the cable connections inside the computer casing once more.

Optical Signal Connections

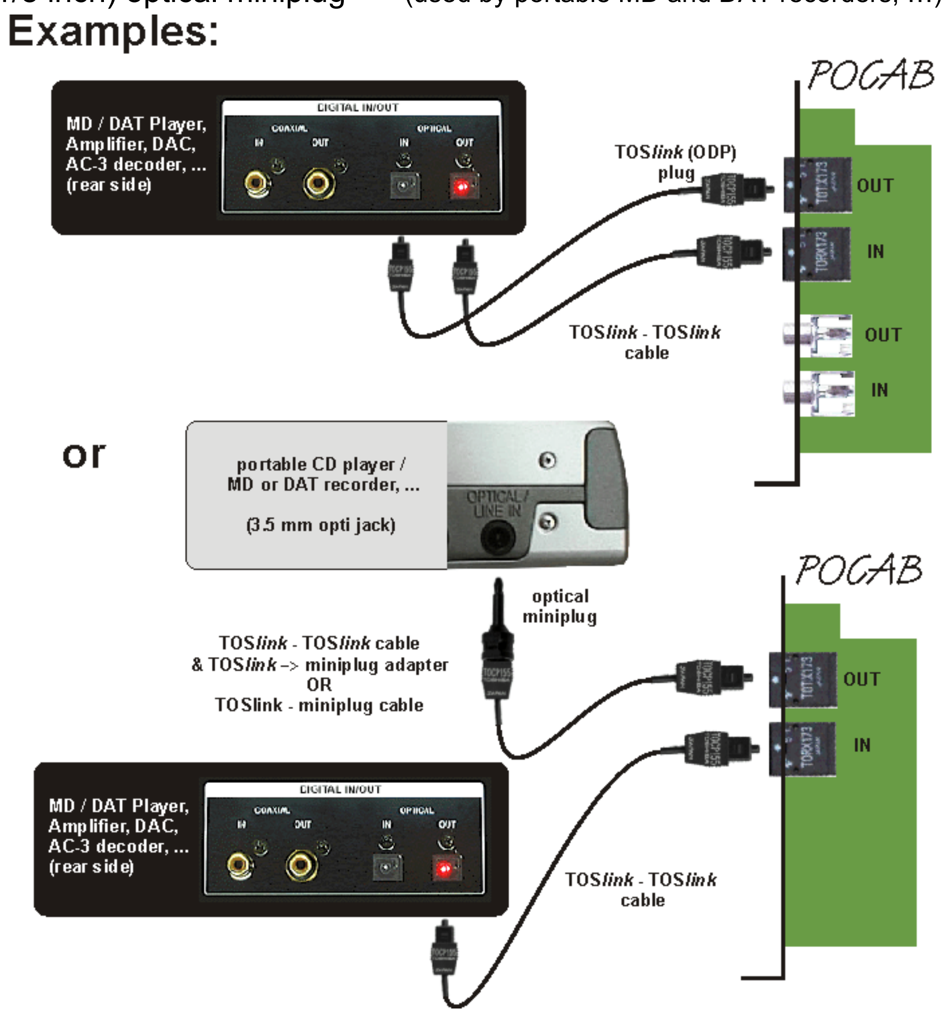
Take TOS*link* (or TOS*link*-to-Miniplug) cable.

TOS*link* cables are offered in nearly arbitrary lengths in most audio shop.

We recommend to limit the length to 10m and to avoid sharp bendings of the cable. Otherwise the cable loses may prevent an error free transmission.

There are mainly two different types of optical plugs:

- ODT (TOS*link*) plug (used by HiFi devices, MD and DAT recorders, *POCAB*, …)
- 3,5mm (1/8 inch) optical miniplug (used by portable MD and DAT recorders, …)



Please use the protective caps for the optical interfaces when they are not in use. This prevents dust scratches etc., which lower the signal quality.

☞ The digital inputs are switched. This means that there is either the optical or the coaxial input signal passed to the SB Audigy/Live!.

Coaxial Signal Connections

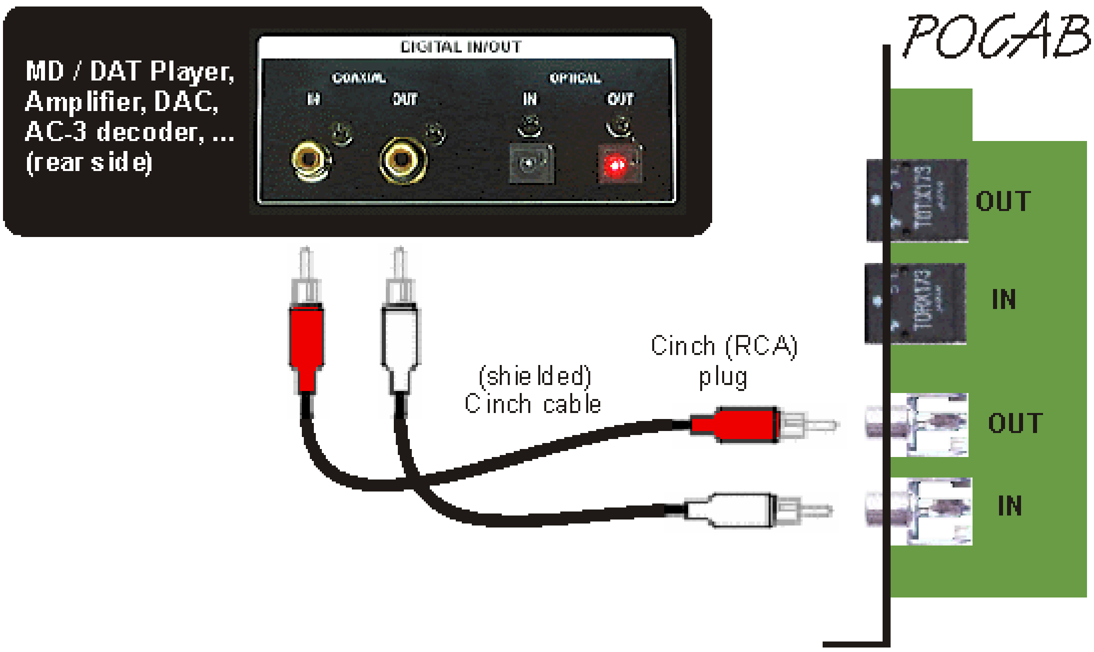
Take cinch (RCA) cable rated at 75 Ohms; usually any cinch (RCA) cable that is good enough to carry a video signal will also be adequate for carrying a digital signal.

☞ For distances of more than 2 meters we recommend to use shielded coaxial cable for reliable and high quality transmission (like “digital coaxial cable”, sometimes also referred to as SPDIF cable).

The input accepts signal levels from 0.5 to ~5..V_{pp}.

The output delivers a voltage level of 1.6 V_{pp}; the signal ground is decoupled from the computer’s ground.

Example:



We tested the input with various signal sources, but we could not test every source. So if you have got some problems, please report them to us, we will try to finger out a solution.

☞ The digital inputs are switched. This means that there is either the optical or the coaxial input signal passed to the SB Audigy/Live!.

Software Setup (Windows 9x, ME, 2000, XP)

Mixer Settings

Start the Creative mixer. It should look similar to our screenshot.

- You will find an SPDIF-IN controller.

The adjustment of the recording level is a little unusual for digital recording. I assume, your sound signals are already digitised, so the volume is fixed. Nevertheless you probably will have to correct the volume at the recording mixer to make sure the signal is not overdriven.

- The SPDIF-OUT signal should be on all the time. It is equivalent to the analog line-out of the SBAudigy/Live!.
  Nevertheless it might be necessary to check “Digital Output Only”, as the same resource is also used for the center speaker.

Please note that every adjustment, e.g. tremble or bass adjustment, has an influence on the digital line-out.

SB Audigy / Live! and Sample Rate questions

The Soundblaster Audigy/Live! (Value) is able to work on all popular sample rates at the input. But note that every signal going into the card will be converted to internal 48 kHz (refering to actual drivers). So the output sample rate is fixed at 48 kHz for SB Live! For SB Audigy the output sample rate can be set to 44.1, 48 or 96 kHz.

For SB Live! the reasons have been given by a Creative scientist at the Soundblaster Live home page *http://www.sblive.com*: (extract)

Dave Rossum, Chief Scientist:

It’s important to remember that the Sound Blaster Live! is much more than just a wavetable synthesizer or a CD playback device. At its heart is the EMU10K1 e‹ects engine, a powerful DSP performing, among other things, mixing of all the various functions in the digital domain. To mix digital audio signals, they must all be at EXACTLY the same sample rate, even deviations of a few parts per million must be eliminated. So when we went to design the EMU10K1, we had to choose a single master sample rate at which the mixer would operate, and of course we had to design sample rate converters to change any incoming audio to match this sample rate. This is the technology required to achieve digital mixing - that’s why it’s first featured by Sound Blaster Live!.

It was obvious that either 44.1 kHz, the CD standard, or 48 kHz, the professional audio and DVD standard, were the only possible choices for the master sample rate. We picked 48 kHz for a variety of reasons. First, if we were processing incoming audio at nominally 48 kHz, use of 44.1 kHz would lose information. Second, even
in preparation of 44.1 kHz CDs in professional studios today, 48 kHz is the preferred standard, with a final conversion to 44.1 kHz as the last step. Third, the nearly twice larger ”guard band” (the difference between 20 kHz and the Nyquist frequency) of 24 kHz means virtually every audio process and effect performs much better at 48 kHz than at 44.1 kHz. And finally, the economical AC-97 CODEC operates at 48 kHz.

The SPDIF outputs provide the exact EMU10K1 effects engine outputs, so they operate at this same 48 kHz rate. So why didn’t we add sample rate converter to each SPDIF output to convert the 48 kHz signal back to 44.1 kHz? The primary reason is cost. Since these outputs are ”master” signals, they should be treated
as very high quality and as such the sample rate converters, to be useful, would be fairly expensive in silicon. Furthermore, since the EMU10K1 is clocked o‹ the AC-97 CODEC master clock which is based on 48 kHz, a separate clock and crystal would be necessary to support 44.1 kHz. At least, if audio quality were important, since a standard phase locked loop or divider system would introduce too much jitter. And since the job could be done externally fairly straightforward, we felt making all users bear the cost burden of this feature for the few who would use it was a poor trade-off. Also, we are trying to promote the uniform system design which will be found in the modern digital studio in which the digital inputs bear the burden of sample rate normalization. For those few users who for some reason seem to need a 44.1 kHz SPDIF output, they’ll either have to purchase a sample rate converter box (Analog Devices makes a 20 chip which will do the job nicely), or (as I would tend to recommend) convert their operation to 48 KHz throughout and if 44.1 kHz is needed for CD, have this jobbed out at CD production like most people do.

Why must all the sound sources pass through the sampling rate converters?

It’s not actually true that every sound source must pass through sample rate conversion. If the Sound Blaster Live! can serve as the master 48 kHz clock for the sound source (as provided for in the newest operating system), then the source need not be converted. But any external source can’t possibly know our EXACT 48 kHz rate, and so its sample rate must be normalized to ours, even if the deviation is very small. This is a fact of digital mixing.

Once again, those most familiar with practices in older digital studios will ask why we didn’t make accommodations for supply and receiving AES black synchronization information to eliminate the necessity of sample rate conversion. The reason is that studio synch is simply too complex an issue for most consumers to grasp - sample rate conversion (at least by E-mu) sounds great and makes the digital patch cord behave just like an analog one.

Specifications

supply current	approx. 50mA
operating voltage	5V, supplied by SB Audigy/Live!
operating temperature	approx. 5°C – 60°C
optical in/out	TOS*link*, 670nm
coaxial input	cinch (RCA) connector, 0.5 .. ~ 5 V_{pp}
coaxial output	cinch (RCA) connector, 1.6 V_{pp}
sample rate, in	2,05 MBit/s (32 kHz, e.g. DSR) 2,82 MBit/s (44.1 kHz, e.g. CD) 3,07 MBit/s (48 kHz, e.g. DAT)
sample rate, out	3,07 MBit/s (48 kHz, e.g. DAT)